FORM PTO-1390 (Modified) (REV 11-98)	U.S. DEPARTMENT OF COMMERCE PATENT AND TRADEMARK OFFICE	ATTORNEY'S DOCKET NUMBER
TRANSMITTAL LETTER TO THE UNITED STATES		**66722-010-7**
DESIGNATED/ELECTED OFFICE (DO/EO/US)		U.S APPLICATION NO. (IF KNOWN, SEE 37 CFR
CONCERNING A FILING UNDER 35 U.S.C. 371		**09/869605**

INTERNATIONAL APPLICATION NO. **PCT/DK/00/00003**	INTERNATIONAL FILING DATE **4 January 2000**	PRIORITY DATE CLAIMED **5 January 1999**

TITLE OF INVENTION
A METHOD AND A DEVICE FOR PROVIDING IMPROVED SPEECH INTELLIGIBILITY

APPLICANT(S) FOR DO/EO/US
HANSEN, Kim, Vejlby

Applicant herewith submits to the United States Designated/Elected Office (DO/EO/US) the following items and other information:

1. ☒ This is a **FIRST** submission of items concerning a filing under 35 U.S.C. 371.

2. ☐ This is a **SECOND** or **SUBSEQUENT** submission of items concerning a filing under 35 U.S.C. 371.

3. ☒ This is an express request to begin national examination procedures (35 U.S.C. 371(f)) at any time rather than delay examination until the expiration of the applicable time limit set in 35 U.S.C. 371(b) and PCT Articles 22 and 39(1).

4. ☒ A proper Demand for International Preliminary Examination was made by the 19th month from the earliest claimed priority date.

5. ☒ A copy of the International Application as filed (35 U.S.C. 371 (c) (2))

    a. ☐ is transmitted herewith (required only if not transmitted by the International Bureau).

    b. ☒ has been transmitted by the International Bureau.

    c. ☐ is not required, as the application was filed in the United States Receiving Office (RO/US).

6. ☐ A translation of the International Application into English (35 U.S.C. 371(c)(2)).

7. ☐ A copy of the International Search Report (PCT/ISA/210).

8. ☐ Amendments to the claims of the International Application under PCT Article 19 (35 U.S.C. 371 (c)(3))

    a. ☐ are transmitted herewith (required only if not transmitted by the International Bureau).

    b. ☐ have been transmitted by the International Bureau.

    c. ☐ have not been made; however, the time limit for making such amendments has NOT expired.

    d. ☐ have not been made and will not be made.

9. ☐ A translation of the amendments to the claims under PCT Article 19 (35 U.S.C. 371(c)(3)).

10. ☐ An oath or declaration of the inventor(s) (35 U.S.C. 371 (c)(4)).

11. ☒ A copy of the International Preliminary Examination Report (PCT/IPEA/409).

12. ☐ A translation of the annexes to the International Preliminary Examination Report under PCT Article 36 (35 U.S.C. 371 (c)(5)).

Items 13 to 20 below concern document(s) or information included:

13. ☐ An Information Disclosure Statement under 37 CFR 1.97 and 1.98.

14. ☐ An assignment document for recording. A separate cover sheet in compliance with 37 CFR 3.28 and 3.31 is included.

15. ☒ A **FIRST** preliminary amendment.

16. ☐ A **SECOND** or **SUBSEQUENT** preliminary amendment.

17. ☐ A substitute specification.

18. ☐ A change of power of attorney and/or address letter.

19. ☐ Certificate of Mailing by Express Mail

20. ☒ Other items or information:

Copy of WO 00/41441, dated 13 July 2000

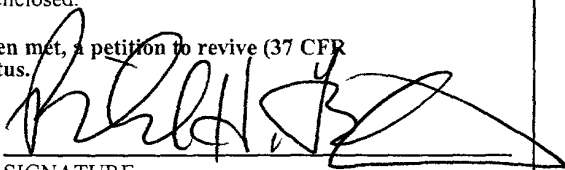| U.S. APPLICATION NO. (IF KNOWN, SEE 37 CFR) 09/869605	INTERNATIONAL APPLICATION NO. PCT/DK/00/00003	ATTORNEY'S DOCKET NUMBER 66722-010-7

21. ` The following fees are submitted:.

BASIC NATIONAL FEE (37 CFR 1.492 (a) (1) - (5)) :

☐ Neither international preliminary examination fee (37 CFR 1.482) nor international search fee (37 CFR 1.445(a)(2) paid to USPTO and International Search Report not prepared by the EPO or JPO $1,000.00

☒ International preliminary examination fee (37 CFR 1.482) not paid to USPTO but Internation Search Report prepared by the EPO or JPO $860.00

☐ International preliminary examination fee (37 CFR 1.482) not paid to USPTO but international search fee (37 CFR 1.445(a)(2)) paid to USPTO $710.00

☐ International preliminary examination fee paid to USPTO (37 CFR 1.482) but all claims did not satisfy provisions of PCT Article 33(1)-(4) $690.00

☐ International preliminary examination fee paid to USPTO (37 CFR 1.482) and all claims satisfied provisions of PCT Article 33(1)-(4) $100.00

CALCULATIONS   PTO USE ONLY

ENTER APPROPRIATE BASIC FEE AMOUNT = $860.00

Surcharge of **$130.00** for furnishing the oath or declaration later than ☐ 20 ☐ 30 months from the earliest claimed priority date (37 CFR 1.492 (e)). $0.00

CLAIMS	NUMBER FILED	NUMBER EXTRA	RATE	
Total claims	6 - 20 =	0	x $18.00	$0.00
Independent claims	3 - 3 =	0	x $80.00	$0.00
Multiple Dependent Claims **(check if applicable)**.			☐	$0.00

TOTAL OF ABOVE CALCULATIONS   = $860.00

Reduction of 1/2 for filing by small entity, if applicable. Verified Small Entity Statement must also be filed (Note 37 CFR 1.9, 1.27, 1.28) **(check if applicable)**. ☐ $0.00

SUBTOTAL   = $860.00

Processing fee of **$130.00** for furnishing the English translation later than ☐ 20 ☐ 30 months from the earliest claimed priority date (37 CFR 1.492 (f)).   + $0.00

TOTAL NATIONAL FEE   = $860.00

Fee for recording the enclosed assignment (37 CFR 1.21(h)). The assignment must be accompanied by an appropriate cover sheet (37 CFR 3.28, 3.31) **(check if applicable)**. ☐ $0.00

TOTAL FEES ENCLOSED   = $860.00

Amount to be: refunded	$
charged	$

☐ A check in the amount of _____ to cover the above fees is enclosed.

☒ Please charge my Deposit Account No. **04-2223** in the amount of **$860.00** to cover the above fees. A duplicate copy of this sheet is enclosed.

☒ The Commissioner is hereby authorized to charge any fees which may be required, or credit any overpayment to Deposit Account No. **04-2223** A duplicate copy of this sheet is enclosed.

NOTE: Where an appropriate time limit under 37 CFR 1.494 or 1.495 has not been met, a petition to revive (37 CFR 1.137(a) or (b)) must be filed and granted to restore the application to pending status.

SEND ALL CORRESPONDENCE TO:

Richard H. Tushin, Esq.
Registration No. 27,297

25269
PATENT TRADEMARK OFFICE

SIGNATURE

Richard H. Tushin
NAME

27,297
REGISTRATION NUMBER

July 2, 2001
DATE

66722-010-7

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re Application of:	)	PATENT
	)	
Kim Vejlby HANSEN	)	Group: Unknown
	)	
Serial No: (PCT/DK00/0003)	)	Examiner: Unknown
	)	
Filed: Concurrently herewith	)	
	)	
A METHOD AND A DEVICE FOR	)	
PROVIDING IMPROVED SPEECH	)	
INTELLIGIBILITY	)	

*   *   *   *   *   *

PRELIMINARY AMENDMENT

Washington, D.C.
July 2, 2001

Honorable Director for Patents
Washington, D.C. 20231

Sir:

Concurrently with the U.S. national phase filing of this

application, please amend this application as follows:

IN THE CLAIMS:

Please amend claim 4 as follows. A marked-up copy of the

claim, showing the changes made thereto, is attached.

4. (Amended) A device according to claim 1, wherein two or

more output signals are produced and where automatic switching

means are provided for switching between the two or more output

signals according to a predetermined scheme.

REMARKS

By this Preliminary Amendment claim 4 has been amended to depend from only one preceding claim. Entry is requested.

Respectfully submitted,

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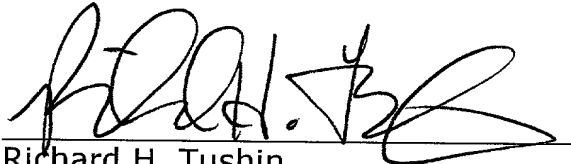RHT/jjr

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PCT/DK00/00003

4. (Amended) A device according to claim 1 [or 2], wherein two or more output signals are produced and where automatic switching means are provided for switching between the two or more output signals according to a predetermined scheme.

TITLE

A method and a device for providing improved speech intelligibility.

The invention relates to a method for the reduction or elimination of an eventual noise contribution to an audio signal, which is picked up by sensor means. The invention further relates to a device for reduction or elimination of

10   an eventual noise contribution to an audio signal, which is picked up by sensor means.

The invention will for practical reasons be explained in connection with a hearing aid. The invention is however not limited to this field as it may be

15   implemented in other technical fields where an audio signal is picked up by sensor means.

Although modern hearing aids have come a long way in providing amplification well suited to the hearing impaired user, many hearing impaired

20   people still have problems understanding speech in noise. The issue of improving the speech intelligibility by improving the ratio between the desired speech signal and the noise has already been addressed in many ways, mostly based on directionality.

25   As of the present day the commercially available directional devices are either an acoustical two-port design as disclosed in US patent no. 5524056, an electrical combination of the outputs of two omni-directional microphones as disclosed in J. Malsano & W. Hottinger, "A method for electronically beam forming acoustical signals and acoustical sensorapparatus", European Patent

30   EP 0 820 210 A2, or an electrical combination of several microphones into a single highly directional device to be used physically externally to the hearing aid as disclosed in W. Soede, A.J. Berkhout & F.A. Bilsen, 1993, "Development

of a directional hearing instrument based on array technology", J. Acoust. Soc. Am. 94(2), p. 785-798.

These previously known devices are thus characterised by a certain amount of directionality pointing in the forward direction with respect to the user, and characterised by a time-invariant processing of the individual ingoing sensor signals. Although the use of directionality to some extent gives improved speech intelligibility, the directionality is in many situations non-satisfactory and there is a requirement of further improving speech inteligibility.

Directionality is also used in a different group of proposed devices, which make use of adaptive noise cancelling principles as disclosed in B. Widrow, J. Glover, J. McCool, J. Kaunitz, C. Williams, R. Hern, J. Zeidler, E. Dong & R. Goodlin, 1975, "Adaptive noise cancelling: Principles and applications", IEEE Proceedings 63, p. 1692-1716. Thus, a number of microphones, which may be mounted either in hearing aids at each ear of the user, on a headband or in a single hearing aid shell, are combined to form estimates of the interfering noise from which the target signal is removed. A signal containing as much target signal as possible is also formed, and these signals are then used as inputs to an adaptive noise canceller. These devices work on the assumption that the target speech signal impinges from a certain predetermined direction, and are characterised by a directionality and a processing of the individual ingoing sensor signals, which is adaptively varying with time. Further, these devices are characterised by the use of the adaptive noise canceller, as disclosed in P.M. Zurek, J.E. Greenberg & P.M. Peterson, 1990, "Adaptive beamforming for noise reduction". United States Patent 4,956,867; P.V.F. Clough & N.A. Lobo, European Patent 0084892; and J. Vanden Berghe & J. Wouters, 1998, "An adaptive noise canceller for hearing aids using two nearby microphones", J. Acoust. Soc. Am. 103(6), p. 3621-3626.

Although the use of adaptive noise cancelling based on directionality to some extend gives improved speech intelligibility, the directionality is in many

situations non-satisfactory in this respect, as it is the case in connection with the use of directionality alone.

Recently, several researchers have published a new principle, which is called

5   independent component analysis (ICA):

1. J. Herault & C. Jutten, "Space or Time adaptive signal processing by neural network models", Neural Networks for Computing, AIP Conference Proceedings 151, Snowbird, Utah, pp. 207-211, 1986.

2. A.J. Bell, United States Patent 5,706,402.

10   3. A.J. Bell & T.J. Sejnowski, "An information-maximisation approach to blind separation and blind deconvolution", Technical Report no. INC-9501, February 1995, Institute for Neural Computation, UCSD, San Diego, CA 92093-0523.

4. T.W. Lee, M. Girolami, A.J. Bell & T.J. Sejnowski, "A unifying information-

15   theoretic framework for independent component analysis", Computers & Mathematics with applications, 1998 in press.

5. T.W. Lee, A.J. Bell & R. Orglmeister, "Blind source separation of real world signals", Proceedings ICNN, USA, 1997.

6. T.W. Lee, A.J. Bell & R. H. Lambert, "Blind separation of delayed and

20   convolved sources", Advances in Neural Information Processing Systems 9, 1997 MIT Press, Camvridge MA pp. 758-764.

7. P. Smaragdis, 1997, "Efficient blind separation of convolved sound mixtures", IEEE Workshop on Applications of Signal Processing to Audio and Acoustics, Mohonk Mountain House, New Paltz, New York.

25   8. K. Torkkola, "Blind separation of delayed sources based on information maximation", Proceedings of the IEEE international conference on acoustic, speech and signal processing, May 7-10 1996, GA, USA.

9. K. Torkkola, "Blind separation of convolved ources based on information maximization", IEEE workshop on neural network for signal processing,

30   Kyoto, Japan, Sept 4-6, 1996.

10. J.F. Cardoso, "Equivariant adaptive source separation", IEEE Transactions on Signal Processing 44(12), 1996.

11.S. Amari, A. Cichocki, H. H. Yang, "A new learning algorithm for blind
signal separation", Advances in neural information Processing systems 8.
MIT Press, 1996.

5   WO 98/25214 discloses a standard ICA method of signal separation where it is
assumed that the sources that generate the sensor signals are stationary. The
same assumption applies to the sensor positions and the source signal
characteristics. In a real world application these assumptions are obviously
violated. WO 98/25214 considers how a standard ICA method should be

10   modified to cope with a non-stationary situation. This comprises a scheme for
choosing the learning rate of the ICA algorithm according to online measures of
the sensor signal characteristics.

US 5,524,056 discloses aspects where the frequency shaping circuit, which

15   ensures that the frequency responses of directional and omni-directional
microphones are similar for frontal incidence of sound. The D-Mic thus
contains an arrangement of microphones similar to the two-sensor
configuration proposed in our patent, but the combination with an ICA
algorithm is not mentioned at all.

20

DE 195 31 388 C1 discloses similarly to WO 98/25214 an extension of a
standard ICA algorithm. In this case an eventual non-linear mixing of the
ingoing source signals is taken into account. Sensor arrangement and the
question of how to make sure that target signal is chosen as the output of the

25   ICA algorithm, are not discussed at all.

US 5,208,786 discloses only second order statistical moments whereas an ICA
algorithm considers higher order statistical moments in the signal separation,
The disclosed method is also known as decorrelation or Principal Component

30   Analysis (PCA). Again, sensor arrangement and the question of how to make
sure that target speech signal is chosen as the output of the algorithm, are not
discussed.

EP 0 565 479 A1 discloses a frequency domain realisation of a standard ICA algorithm. The sensor arrangement and the question of how to make sure that target signal is chosen as the output of the ICA algorithm, are not discussed.

5

US 5,706,402 discloses what has become standard ICA. Thus it outlines the technical background of the present invention. The sensor arrangement and the question of how to make sure that target signal is chosen as the output of the ICA algorithm are not discussed,

10

Paper XP-002108812 discloses an experimental application of a standard ICA algorithm in a laboratory setting with speech and noise signals in an anechoic stationary environment. The sensor arrangement is two directional microphones placed on top of each other facing in directions 90 degrees apart. No explanation for the particular choice of sensor configuration is given, and the method with which a particular output signal from the ICA signal separation algorithm has been designated as the target speech signal is not discussed at all.

20 The objective of ICA is to recover the underlying independent source signals given only sensor observations that are linear mixtures of the original source signals. The only assumption of ICA is that the original source signals are statistically independent, otherwise the statistics of the source signals and the mixing of these into the sensor signals may be unknown. In contrast to correlation-based transformations such as Principal Component Analysis (PCA, I.T. Jolliffe, Principal Component Analysis, 1986, Springer Verlag), which decorrelates signals according to 2nd-order statistics, ICA also reduces higher-order statistical dependencies, in terms of maximising joint output entropy, in order to extract statistically independent signal components.

30

In the linear blind signal separation problem, N signals, $\mathbf{s}(t) = [s_1(t),...,s_N(t)]^T$, are mixed so that an array of N sensors picks up a set of signals $\mathbf{x}(t) = [x_1(t),...,x_N(t)]^T$, each of which has been mixed, delayed and filtered as follows

$$x_i(t) = \sum_{j=1}^{N} \sum_{k=0}^{M-1} a_{ijk} s_j(t - D_{ij} - k)$$

where D_{ij} are entries in a matrix of delays and $\mathbf{a}_{ij}$ are the M - tap filter coefficients between the jth source and the ith sensor. The problem is to invert this scrambling without knowledge of it, thus recovering the original signals $\mathbf{s}(t)$ given only the $\mathbf{x}(t)$ signals. Finding this inverse scrambling is a challenging task since no informations are provided about the mixing nor the signals (hence the term blind separation). The type of architecture chosen for inverting the scrambling is important and can be made in numerous ways. An accurate architecture for inverting a M-tap filter is an infinite impulse response (IIR) filter with M coefficients. However, IIR filters are limited to have poles inside the unit circle, which imply that a stable filter only exists for a minimum phase system. FIR filters may be used to approximate the inverse solution. Thus the inverse scrambling is performed according to

$$u_i(t) = \sum_{j=1}^{N} \sum_{k=0}^{M-1} w_{ijk} x_j(t - d_{ij} - k)$$

which has filters, w_{ij}, and delays d_{ij}, which supposedly reproduce, at the output $\mathbf{u}(t)$, the original uncorrupted source signals, $\mathbf{s}(t)$, apart from a scaling factor for each signal and a permutation of signals.

Several algorithms have been proposed for the blind separation of linear mixtures. Bell and Sejnowski (3) proposed to learn the separating process by minimising the mutual information between components of $\mathbf{y}(t) = g(\mathbf{u}(t))$, where g is a non-linear function approximating the cumulative probability density function of the sources. They showed that for positively kurtotic

signals (like speech) minimising the mutual information between components of $y(t)$ is equal to maximising the entropy of $y(t)$, which can be written as $H(\mathbf{y}) = -E\left[\ln(f_y(\mathbf{y}))\right]$, where $f_y(\mathbf{y})$ denotes the probability density function of $y(t)$. Denoting the determinant of the Jacobian of the whole unmixing process by $|J|$, $f_y(\mathbf{y})$ can be written as $f_x(\mathbf{x})/|J|$ (the Jacobian is a matrix with entries of $\partial y_i / \partial x_j$). Maximising the entropy of the output leads to maximising $E\left[\ln(|J|)\right]$, which in turn can be developed into a stochastic gradient ascent rule using instances of $x(t)$ and $y(t)$, instead of using the expectation. Thus

$$\Delta \mathbf{W} \propto (1 - 2\mathbf{y}(t))\mathbf{x}(t)^T + \left[\mathbf{W}^T\right]^{-1}$$

where $g(u) = 1/(1 + e^{-u})$ is used to approximate the cdf.

The algorithm can be made more efficient and independent of the conditioning of the mixing process (matrix) by using the so-called natural gradient instead of the absolute gradient, see Amari (11).

One particular proposed application of ICA is within electroencephalographic (EEG) recording of scalp potentials in humans and related brain activity measurements.

The application of the ICA to hearing aids has also been mentioned in H. Sahlin, "Blind signal separation by second order statistics", Ph.D. thesis, Chalmers University of Technology, Technical Report no. 345, Sweden, 1998, but it has so far never been suggested how the implementation of this technique could be realised in connection with audio systems in order to improve speech intelligibility.

Based on this prior art the objective of the present invention is to provide a method and a device for reducing noise in an audio signal comprising both noise and target signal, which method and device has an increased

functionality and reliability compared with the prior art within the audio field of technology.

SUMMARY OF THE INVENTION

5

This is achieved by means of a method comprising the steps of:

providing at least two input signals, each having different contents of target signal and noise;

processing the two input signals;

10   the processing comprising use of an independent component analysis or a similar technique based on the differences of the at least two input signals, hereby determining whether statistical dependent signal elements are present and removing at least part of unwanted signal elements;

outputting a part of the audio signal.

15

The target signal is defined as the signal coming from in front of the device performing the method, whereas other signals are considered as noise signals. In case of no specific well-defined source signal and cases laying outside the assumed set-up, the noise elimination will disappear meaning that the signal

20   processing strategy will pass the input signals unaltered to the outputs.

In a preferred embodiment the method comprises at least two input signals, which are picked up at least at two mutually distanced locations. Hereby one signal contains the target signal with a higher signal-to-noise ratio than the

25   other input signal. In this case the sensor may be identical.

In a further preferred embodiment the method comprises at least two input signals, which are based on differences in the directionality, preferably a directional and an omni-directional sensor. Hereby it becomes possible to

30   determine the origin of the respective signal components based on the differences of the directionality applied when sensing the two input signals.

The different embodiments show that the differences in signal-to-noise ratio is based on the actual situation of use, which means that it is the mutual differences in signal-to-noise ratio in relation to a desired target signal, which is relevant, and not the signal-to-noise ratio of the sensor means itself.

In a further preferred embodiment two or more output signals are produced and where a possibility exists for switching between the two or more output signals or combinations of these. Alternatively an automatic switching between the two or more output signals according to a predetermined scheme is provided. Hereby it becomes possible to make a choice of the actual influence of the separation method on the noise canceling in the actual situation or at least have a change according to the actual choice of the predetermined scheme, which may be adapted for switching at different listening environments.

The objective of the invention is further achieved by means of a device comprising:

at least two input channels;

signal processing means in connection with the input channels, which provides input signals to the signal processing means;

a receiver in connection with the signal processing means;

the signal processing means being adapted to process the signals by means of an independent component analysis method or a similar method based on differences of signal-to-noise ratios of the input signals in relation to a desired target signal, the processing comprising determining whether statistical dependent signal elements are present and removing at least part of the unwanted signal elements, thereby enhancing other parts of the audio signal.

In a preferred embodiment the device comprises two sensors, preferably microphones, having different signal-to-noise ratio (S/N-ratio). One of the sensors is chosen as the target sensor. Hereby it becomes possible to separate

the respective signal components based on the noise content of the further sensor.

In a preferred embodiment the device comprises a directional microphone and an omni-directional microphone. Hereby the directional microphone will preferably contain the desired output signal based on the position of the user facing the desired audio signal source.

In a preferred embodiment the two microphones are mutually distanced. Hereby it becomes possible to determine the origin of the respective signal components based on a time delay in reception of the two input signals. This makes it possible to make a beamforming of the input signals hereby possibly adding directionality to at least one of the inputs. Other ways of beamforming may be used in this connection.

In a preferred embodiment two or more output signals are produced and means are provided for switching between the two or more output signals or combinations of these. Alternatively means are provided for automatic switching between the two or more output signals according to a predetermined scheme.

Due to the fact that most hearing impaired have a hearing disorder which makes it even more difficult than for normal hearing persons to separate a target signal from the noise, which is often present in a speech situation, the invention is particularly relevant in connection with the technical field of hearing aids.

The invention therefor further relates to a hearing aid comprising: at least two microphones for audio signal input; signal processing means in connection with the microphones; an amplifier in connection with the signal processing means; a receiver in connection with the amplifier for outputting a signal from the amplifier; the signal processing means being adapted to process the signals

by means of an independent component analysis method or a similar method based on the input from the at least two microphones, the processing comprising determining whether statistical dependent signal elements are present and removing at least part of the unwanted signal elements, thereby

5   enhancing other parts of the audio signal.

The hearing aid according to the invention may further comprise the features set forth above, either separate or in combination.

10   Other fields of relevant use of the invention may be telecommunication or audio systems. In such systems the input and output may be connected to antennas or similar transmission and receiving means or may comprise microphones as input means as in the case of a hearing aid. Other elements of such systems may be standard elements, as these are not influenced by the

15   signal processing according to the invention.

The invention will be described more detailed with reference to the accompanying drawings.

20   BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1 is a diagram showing the principles of the invention;

Fig. 2 is a schematic diagram showing the principles of an implemented version of the invention;

25   Fig. 3 is a schematic diagram showing the principles of an implemented version of the invention;

Fig. 4 is a schematic diagram showing the implemented version of the invention as shown in fig. 2 further implemented in a hearing aid.

30   DESCRIPTION OF THE PREFERRED EMBODIMENT

The fundamental principle of the invention is schematically shown in fig. 1. The invention is basically a system, e.g. a hearing aid, with two or more sensors and a calculation unit. The calculation unit carries out the separation of the target and noise signals, by using the independence of the mixed signals

5 according to ICA or a similar method comprising the basics of the ICA. The sensors are arranged so that one is positioned to receive sound primarily from a target direction in front, whereas the others have arbitrary characteristics that do not specifically favour the target direction. Hereby, it is possible to use the technique of independent component analysis to separate the desired signal,

10 which impinges from the target direction, from the disturbing noise signals, which impinge from any other directions.

To illustrate the invention, an example is given of a system implementing signal processing as described above. Fig. 2 schematically shows the signal

15 processing system. The system comprises a directional and an omni-directional microphone, and a digital signal processing unit implementing the signal separation algorithm. Using the directional microphone gives the target direction from in front of the user, whereas the omni-directional microphone gives a signal equally representing all signals around the head of the user.

20

A particularly important property of the independent component analysis is that it separates convolved and delayed source signals, where each independent source signal is defined as a signal which appears in the same way within each mixing process. Another important characteristic about the

25 independent component analysis is that knowledge about the ratio of the source signals within the mixed signals can be used for classifying the separated signals. If for instance one source signal appears with a significantly better signal to noise ratio in one of the sensor signals, this information can be used to ensure that this source signal always will appear in a fixed output.

30 Within the present invention these two characteristics combined with an appropriate placement of at least two sensors are exploited to eliminate signals not coming from in front of the user of the device.

From fig. 3 an embodiment appears, which comprises the features of the embodiment of fig. 2, but where the signal processor produces two output signals. By means of switching means one of the two output signals may be

5 selected for further processing, e.g. amplification, or for output.

From fig. 4 an embodiment appears as a hearing aid according to the invention. The essential components of the hearing aid comprise two microphones, preferably a directional microphone and an omni-directional microphone, and

10 an A/D converter connected to each of the microphones. The A/D converters are connected to a digital signal processor, which is adapted to perform the ICA method on the incoming signals. The signal from the signal processor is then lead to an amplifier and from this through a D/A converter to a receiver for performing the output of the processed signal. The devices of the figs. is in a

15 usual manner powered by means of usual power sources, such as batteries.

CLAIMS

1.   A device for use in reducing noise in an audio signal containing noise and a target signal, the device comprising:

at least two input channels each receiving a signal from a respective microphone, at least two of the microphones having mutually different directionality;

signal processing means in connection with the input channels, which provides input signals to the signal processing means;

a receiver in connection with the signal processing means;

the signal processing means being adapted to process the signals by means of an independent component analysis method based on differences of signal-to-noise ratios of the inputs signals in relation to a desired target signal, the processing comprising determining whether statistical dependent signal elements are present and removing at least part of the unwanted signal elements, thereby enhancing other parts of the audio signal.

2.   A device according to claim 1, wherein the device comprises at least a directional microphone and an omni-directional microphone.

3.   A device according to claim 1, wherein two or more output signals are produced and where means are provided for switching between the two or more output signals or combinations of these.

4.   A device according to claim 1 or 2, wherein two or more output signals are produced and where automatic switching means are provided for switching between the two or more output signals according to a predetermined scheme.

5.   A device for use in reducing noise in an audio signal containing noise and a target signal, the device comprising:

at least two microphones and beamforming means to make a beamforming of the input signals from the microphones and hereby adding directionality to the signal of at least one of at least two input channels;

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29. 03. 2001

29-03-2001

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WO 00/41441   15   PCT/DK00/00003
Amended claims March 26, 2001.

signal processing means in connection with the input channels, which provides input signals to the signal processing means;

a receiver in connection with the signal processing means;

the signal processing means being adapted to process the signals by means of an independent component analysis method based on differences of signal-to-noise ratios of the inputs signals in relation to a desired target signal, the processing comprising determining whether statistical dependent signal elements are present and removing at least part of the unwanted signal elements, thereby enhancing other parts of the audio signal.

6.   A hearing aid comprising at least two microphones each having a different signal-to-noise ratio in relation to a desired target signal; signal processing means in connection with the at least two microphones; an amplifier in connection with the signal processing means; a receiver in connection with the amplifier for outputting a signal from the amplifier; the signal processing means being adapted to process the signals by means of an independent component analysis method based on the input from the two microphones having a mutually different signal-to-noise ratio in relation to a desired target signal, the processing comprising determining whether statistical dependent signal elements are present and removing at least part of the unwanted signal elements, thereby enhancing other parts of the audio signal, wherein the hearing aid comprises a directional microphone and an omni-directional microphone.
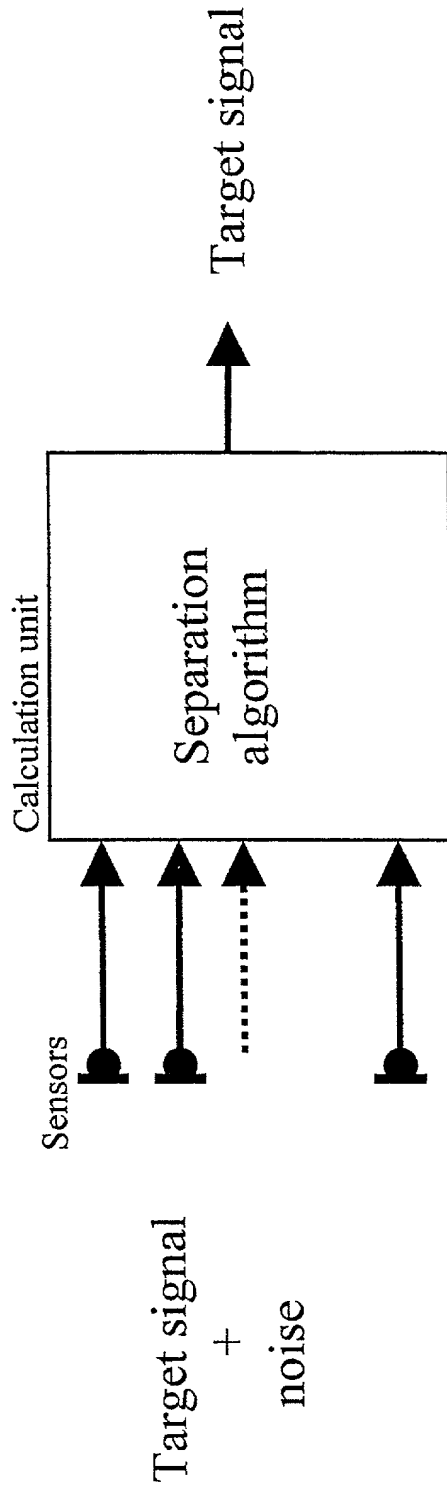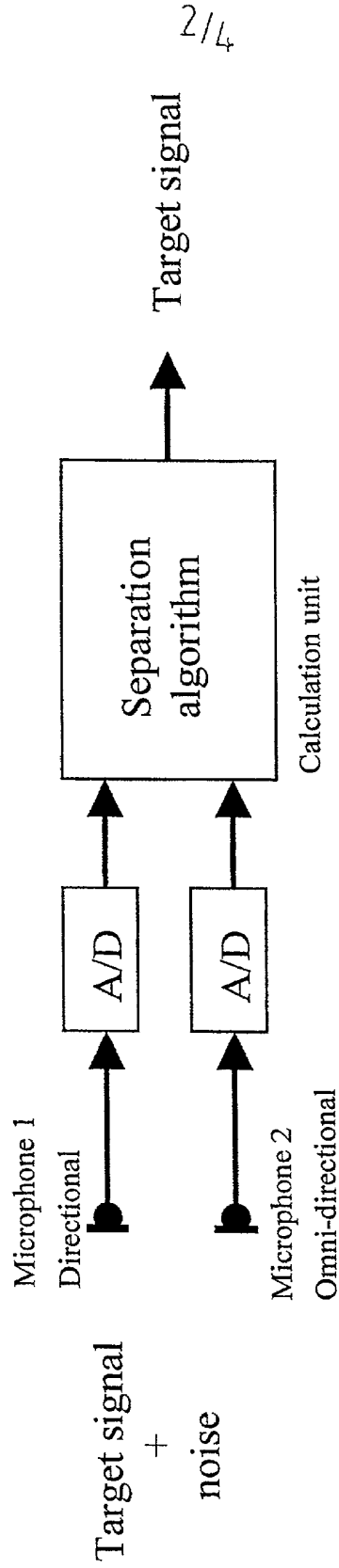
AMENDED SHEET

1/4
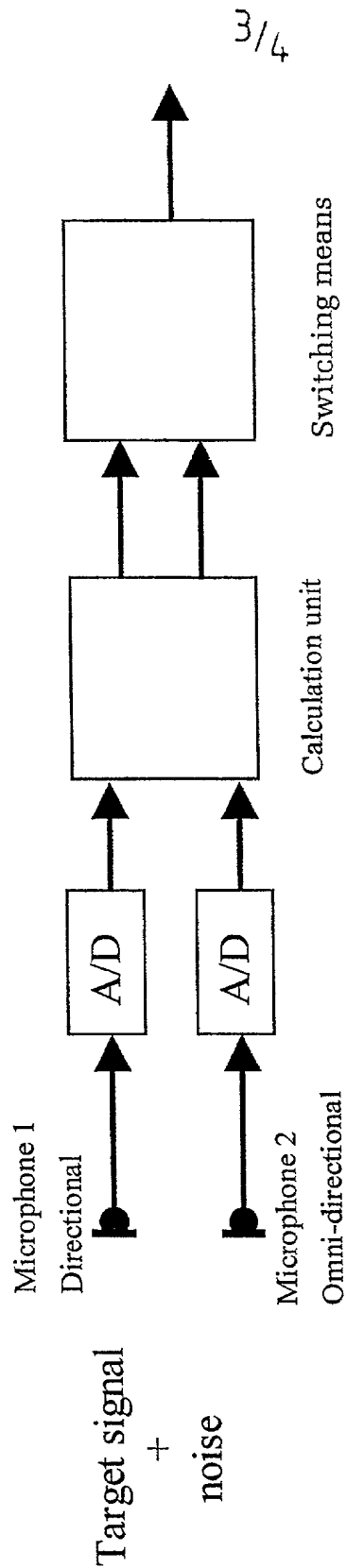

FIG. 1

2/4

Target signal

Target signal
+
noise

Microphone 1
Directional

Microphone 2
Omni-directional

A/D

A/D

Separation
algorithm

Calculation unit

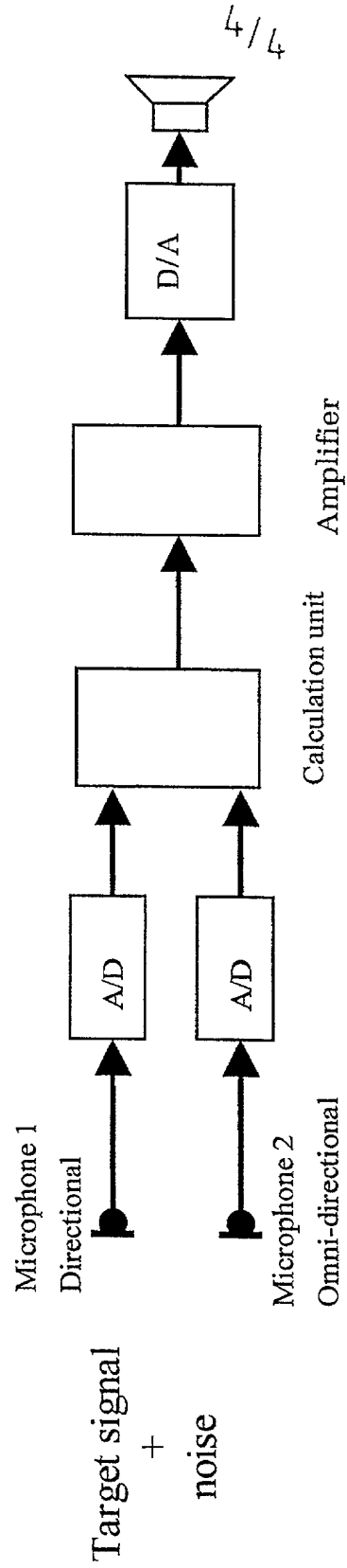**FIG. 2**

3/4

FIG. 3

FIG. 4

COMBINED DECLARATION AND POWER OF ATTORNEY FOR **UTILITY** PATENT APPLICATION (Includes PCT)	Attorney Docket No. 66722-

As a below named inventor, I hereby declare that:
My residence, post office address and citizenship are as stated below next to my name; that

I believe I am the original, first and sole inventor (if only one name is listed below) or an original, first and joint inventor (if plural inventors are listed below) of the subject matter which is claimed and for which a patent is sought on the invention entitled:

A METHOD AND A DEVICE FOR PROVIDING IMPROVED SPEECH INTELLIGIBILITY

the specification of which (check one):   [ ] is attached hereto.

[ ] was filed on _____ as Application Serial No. _____ and was amended on _____.

[ ] was filed as PCT international application no. PCT/DK00/00003 on 4 January 2000 and was amended under PCT Article 19 on _____ (if applicable).

I hereby state that I have reviewed and understand the contents of the above-identified specification, including the claims, as amended by any amendment referred to above.

I acknowledge the duty to disclose information which is material to the examination of this application in accordance with Title 37, Code of Federal Regulations, §1.56(a).

I do not know and do not believe the claimed invention was ever known or used in the United States of America before my or our invention thereof, or patented or described in any printed publication in any country before my or our invention thereof or more than one year prior to this application, that the same was not in public use or on sale in the United States of America more than one year prior to this application, that the invention has not been patented or made the subject of an inventor's certificate issued before the date of this application in any country foreign to the United States of America on an application filed by me or my legal representatives or assigns more than twelve months prior to this application.

I hereby claim foreign priority benefits under Title 35, United States Code §119 of any foreign application(s) for patent or inventor's certificate listed below and have also identified below any foreign application for patent or inventor's certificate having a filing date before that of the application(s) on which priority is claimed:

Prior Foreign Application(s)   Priority Claimed

99610002.0	European	5 January 1999	[X] [ ]
(Number)	(Country)	Day/Month/Year Filed	Yes No
			[ ] [ ]
(Number)	(Country)	Day/Month/Year Filed	Yes No
			[ ] [ ]
(Number)	(Country)	Day/Month/Year Filed	Yes No

I hereby claim the benefit under Title 35, United States Code, §119 (e) of any United States provisional application(s) listed below:

Application No.	Day/Month/Year Filed	Application No.	Day/Month/Year Filed

I hereby claim the benefit under Title 35, United States Code, §120 of any United States application(s) or PCT international application(s) designating the United States of America listed below and, insofar as the subject matter of each of the claims of this application is not disclosed in the prior application(s) in the manner provided by the first paragraph of Title 35, United States Code, §112, I acknowledge the duty to disclose material information as defined in Title 37, Code of Federal Regulations, §1.56(a) which occurred between the filing date of the prior application and the national or PCT international filing date of this application:

| Application Serial No. | Filing Date | Status (patented, pending, abandoned) |

| Application Serial No. | Filing Date | Status (patented, pending, abandoned) |

I hereby appoint the following attorney(s) and/or agent(s) to prosecute this application and to transact all business in the Patent and Trademark Office connected therewith: Lawrence R. Radanovic, Reg. No. 23,077; Richard H. Tushin, Reg. No. 27,297; Donald N. Huff, Reg. No. 27,561; John P. DeLuca, Reg. No. 25,505; Sandra S. Snapp, Reg. No. 41,444; Charles Rutherford, Reg. No. 18,933; Robert L. Kelly, Reg. No. 31,843; Ernest E. Helms, Reg. No. 29,721; and William F. Kolakowski, Reg. No. 41,908, all of Dykema Gossett PLLC. Direct all telephone calls to telephone no. (202) 522-8600 and faxes to (202) 522-8669.

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I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true, and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful false statements may jeopardize the validity of the application or any patent issued thereon.

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